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Docket No.: 290627US8PCT

COMMISSIONER FOR PATENTS
ALEXANDRIA, VIRGINIA 22313

RE: Application Serial No.: 10/500,158
Applicants: Craig George COCKERTON
Filing Date: January 27, 2005
For: AUDIO VISUAL MEDIA ENCODING SYSTEM
Group Art Unit: 2614
Examiner: Ramakrishnaiah, M.

SIR:

Attached hereto for filing are the following papers:

Petition to Make Special, Information Disclosure Statement, International Preliminary Examination Report

Our credit card payment form in the amount of **$130.00** is attached covering any required fees. In the event any variance exists between the amount enclosed and the Patent Office charges for filing the above-noted documents, including any fees required under 37 C.F.R 1.136 for any necessary Extension of Time to make the filing of the attached documents timely, please charge or credit the difference to our Deposit Account No. 15-0030. Further, if these papers are not considered timely filed, then a petition is hereby made under 37 C.F.R. 1.136 for the necessary extension of time. A duplicate copy of this sheet is enclosed.

Respectfully submitted,

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IN THE UNITED STATES PATENT & TRADEMARK OFFICE

IN RE APPLICATION OF	:
CRAIG GEORGE COCKERTON	: EXAMINER: RAMAKRISHNAIAH, M.
SERIAL NO: 10/500,158	:
FILED: JANUARY 27, 2005	: GROUP ART UNIT: 2614
FOR: AUDIO VISUAL MEDIA ENCODING SYSTEM	:

PETITION TO MAKE SPECIAL

COMMISSIONER FOR PATENTS
ALEXANDRIA, VIRGINIA 22313

SIR:

I. Basis for the Petition

Pursuant to MPEP §708.02(VIII) (8th ed. Revised August 2005), Applicants hereby petition for a special status for this Application.

II. Requirements for Granting Special Status

MPEP §708.02(VIII) established five requirements for a grant of special status. The following subsections show that each of these five requirements is satisfied in the above-identified case.

08/25/2006 TBESHAH1 00000064 10500158
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130.00 OP

A. Submit Petition and Fee: §708.01(VIII)(A)

This petition is accompanied by the fee set forth in 37 C.F.R. §1.17(h).

B. Agree to Election Without Traverse: §708.02(VIII)(B)

Applicants submit that Claims 37-69 are directed to a single, patentable invention. If a restriction requirement is imposed in this Application, Applicants agree to make an election without traverse.

C. State that a Preexamination Search was Made: §708.02(VIII)(C)

A pre-examination search has been made by one or more foreign patent offices in corresponding foreign applications having claims of the same or similar scope to the claims in this application. An International Search was conducted in corresponding PCT application PCT/NZ 03/00187. A copy of the PCT Search Report has been filed. Furthermore, Information Disclosure Statements were filed on January 27, 2005 and July 28, 2006. The pending claims are substantially similar to the claims searched in the corresponding PCT application.

D. Submit a Copy of the Most Relevant References: §708.02(VIII)(D)

All references identified in the prior art search have been made of record in the Information Disclosure Statement filed January 27, 2005 and July 28, 2006. All references now of record are discussed below with reference to the claimed subject matter.

E. Submit a Detailed Discussion of the References, Pointing Out How the Claimed Subject Matter is Patentable Over the References: §708.02(VIII)(E)

Consistent with the searches discussed above, Applicants respectfully submit that the claims of the Application patentably distinguish over all references now of

record. A detailed discussion pursuant to 37 C.F.R. §1.111 is provided below for pointing out with particularity how the claimed subject matter is patentable over the references of record.

Applicant's Claim 37, recites

> A method of encoding audio visual media signals characterised by the steps of:
>
> (i) receiving a videoconference transmission from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals, and
>
> (iii) applying a selected encoding process to a received audio visual signal, said encoding process being selected depending on the contents of said at least one protocol signal read.

Applicant's Claim 63 recites

> A method of encoding audio visual media signals characterised by the steps of:
>
> (i) receiving a videoconference transmission from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals, and
>
> (iii) determining the time position of a keyframe present within an audio visual signal received, and
>
> (iv) encoding a keyframe into the encoded output at the same time position at which the keyframe was detected in the original received audio visual signal.

Applicant's Claim 64 recites

> A method of encoding audio visual media signals, characterised by the steps of:
>
> (i) receiving a videoconference transmission from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals, and
>
> (iii) detecting a content switch within the audio visual content of a received audio visual signal or signals, and
>
> (iv) encoding an index marker at the time position at which the content switch was detected.

Applicant's claim 66 recites

> A method of encoding audio visual media signals characterised by the steps of:
>
> (i) receiving a videoconference transmission, from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals, and
>
> (iii) detecting a content switch within the audio visual content of a received audio visual signal, and
>
> (iv) encoding a keyframe and
>
> (v) encoding an index marker at the same time position or adjacent to the time position of the keyframe encoded.

Applicant's Claim 67 recites

> A method of encoding audio visual media signals characterised by the steps of:

> (i) receiving a videoconference transmission from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals, and
>
> (iii) detecting the existence of a low content state present within a received audio visual signal or signals, and
>
> (iv) time compressing the encoded output content during the time period in which said low content state is detected within the videoconference transmission received.

Applicant's Claims 69 recites

> A method of encoding audio visual media signals characterised by the steps of:
>
> (i) receiving a videoconference transmission from a computer network, said videoconference transmission including at least one audio visual signal and at least one protocol signal, and
>
> (ii) reading one or more protocol signals to determine the encoding characteristics of the received videoconference transmission,
>
> (iii) receiving encoding preferences from at least one user, and
>
> (iv) selecting from a set of encoding processes a subset of encoding processes which can be implemented using the user's preferences and the encoding characteristics, and
>
> (v) displaying the subset of encoding processes to a user.

U.S. Patent No. 6,466,248 (Spann et al., hereinafter Spann) is directed toward a technique of recording a videoconference in which the videoconference can be

recorded independently of the real time transmission of the video signal.[1] The video conferencing system includes a first encoder provided for encoding the original video signal into a first format suitable for streaming transmission, and a second encoder provided for encoding the original video signal into a second format having a higher quality or less storage as compared to the first format.[2] The first and second encoders work independently of each other. The first encoder uses a low latency algorithm for real time transmission of the video signal. This encoder sacrifices some quality to achieve real time transmission with low latency within an allocated bandwidth. The second encoder is used for archival and storage of the image.[3]

Furthermore, WO/01/178393 is a PCT application that corresponds to Spann, EP1290881 is a European Application that corresponds to Spann, JP 2003530787 is a Japanese Application that corresponds to Spann, and CN 1421099 is a Chinese Application that corresponds to Spann. Accordingly, Applicants respectfully submit that the discussion of Spann applies to WO/01/178393, EP 1290881, JP 2003530787, and CN 1421099.

The Information Disclosure Statement filed on January 27, 2005 lists 5 patents for Yurt et al. These patents are all related to each other. As explained on the face of U.S. Patent No. 6,144,702, 6,144,70 is a division of U.S. Patent No. 6,002,720, which is a continuation of U.S. Patent No. 5,550,863, which is a continuation of U.S. Patent No. 5,253,275, which is a continuation of U.S. Patent No. 5,132,992. Furthermore, W0/92/12599 is PCT application that corresponds to U.S. Patent No. 5,132,992. Accordingly, these references will be discussed together, with citations referring to U.S. Patent No. 6,144,702 (hereinafter Yurt).

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[1] Spann, col. 2, lines 6-9.
[2] Spann, abstract.
[3] Spann, col. 2, lines 20-24.

Yurt is directed toward a system of distributing video and/or audio information employing digital signal processing to achieve high rates of data compression.[4]

Yurt discloses that audio data is preferably compressed by audio compressor 128 by application of an adaptive differential pulse code modulation (ADPCM) process to the audio data. Compression by compressor 116 may be performed on a group of 24 video frames may preferably be passed in sequence to the frame buffer 130 of the video precompression processor 115b where they are analyzed by video compressor 129 which performs data reduction processing on the video data. Video compression is preferably performed by video compressor 129. Video compression is achieved by the use of processors running algorithms designed to provide the greatest amount of data compression possible. Video data compression preferably involves applying two processes: a discrete cosine transform, and motion compensation. [5]

U.S. Patent No. 5,764,277 (Loui et al., hereinafter Loui) is directed toward a method and apparatus for coded-domain combining of video signals from multiple video conference users into a combined video signal suitable for transmission to all of the users.[6]

Fig. 5 of Loui shows a video conferencing system 100, which includes GOB-based coded-domain video signal combiner 105. A plurality N of video conference system users 110-i, i=1, 2 . . . N (where N.ltoreq.12 in this example), transmit video signals to corresponding buffers 120-i in the combiner 105. The user's 110-i transmit coded digital video signals at a data rate R via conventional codecs. The buffers 120-i store one or more of the incoming video frames from each of the users until the GOBs

[4] Yurt, abstract.
[5] Yurt, col. 9, lines 40-65.
[6] Loui, col. 3, lines 33-36.

thereof are needed for insertion into an output frame in the manner previously described in conjunction with Fig. 4. The buffers 120-i also serve to accommodate variable-length coded GOBs. The buffers 120-i supply the corresponding video signals to a GOB multiplexer 130 which processes the incoming signals to provide an output video signal including combined output frames such as frame 30 of Fig. 4. Multiplexer 130 is operative to identify the GOBs in the incoming video streams from each of the N users 110-i and to multiplex the incoming GOBs into a single output CIF video stream.[7]

Furthermore, the reference on line AA of the IDS filed on July 28, 2006 (Sun Ming-Ting et al.) is Loui (U.S. Patent No. 5,764,277). This reference was inadvertently listed a second time.

U.S. Patent Publication No. 2001/0047517 (Christopolous et al., hereinafter Christopolous) is directed toward a method and apparatus is described for performing intelligent transcoding of multimedia data between two or more network elements in a client-server or client-to-client service provision environment. Accordingly, one or more transcoding hints associated with the multimedia data may be stored at a network element and transmitted from one network elements to another. One or more capabilities associated with one of the network elements may be obtained and transcoding may be performed using the transcoding hints and the obtained capabilities in a manner suited to the capabilities of the network element.[8] Fig. 1 of Christopoulos illustrates various network components for the communication of multimedia data. The network includes a server 110, a gateway 120 and client 130. Server 110 stores multimedia data, along with transcoding hints, in multimedia

[7] Loui, col. 5, lines 31-52.
[8] Christopoulos, abstract.

storage element 113. Server 110 communicates the multimedia data and the transcoder hints to gateway 120 via bidirectional communication link 115. Gateway 120 includes a transcoder 125. Transcoder 125 reformats the multimedia data using the transcoder hints based upon client capabilities, user preferences, link characteristics and/or network characteristics. The transcoded multimedia data is provided to client 135 via bidirectional communication link 130.[9]

EP 1069779 (Kitamura) is directed toward a transcoder for executing a re-coding process on an encoded stream generated based on an MPEG standard in order to generate a re-coded stream having a different GOP (Group of Pictures) structure or bit rate. Specifically, a decoding device of a transcoder 106 decodes a source encoded stream to generate decoded video data and extracts past coding parameters superposed in the encoded stream as history_stream(). In this case, the decoding device extracts the past coding parameters based on information superposed in the encoded stream as re_coding_stream_info(). An encoding device receives the decoded video data and the past coding parameters and uses the past coding parameters to carry out an encoding process in a manner such that this process will not degrade image quality, thereby generating a re-coded stream. Further, the encoding device selects one of the past coding parameters which are optimal for an application connectively following the encoding device and describes only the selected past coding parameters in the encoded stream as history_stream(). The encoding device superposes, as re_coding_stream_info(), information indicating the selected past coding parameters so that the following application can properly extract the coding parameters for the history_stream() from the re-coded stream.[10]

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[9] Chrisyopoulos, paragraph [0035].
[10] Kitamura, abstract.

Kitamura discloses that television conference systems that moving image signals to remote sites use line interframe correlationship among video signals to compressive-code image signals in order to efficiently used transmission paths.[11] Figs. 14 and 15 of Kitamura show the configuration of a transcoder 101. Transocoder 101 converts the GOP structure and bit rate of an encoded video bit stream input to a decoding device102. In order to vary the GOP structure and bit rate of a bit stream, the first, second, third and fourth transcoders are connected in series.[12] Within transcoder 101, the third encoded stream has described therein the third encoding parameters as well as the first- and second-generation encoding parameters generated during the first and second encoding process, respectively.[13] The first- and second-generation encoding parameters are described in a user data area of the picture layer of the third-generation encoded video stream as a history stream.[14] The history information-separating device 105, shown in Fig. 15 of Kitamura, extracts the baseband video data from the transmitted data, supplies this data to encoding device 106, executes the first-, second-, and third-generation history information from the transmitted data, and supplies this information to encoding device 106.[15]

U.S. Patent No. 5,764,901 (Skarbo et al., hereinafter Skarbo) is directed toward a method, apparatus, and storage medium for processing data streams of a data conference between a local user and a remote user. The data conference includes at least one remote video source associated with the at least one remote user and a local video source associated with the local user. A dialog window is displayed for the local user, the dialog window having a record control. One of the at least one

[11] Kitamura, paragraph [0025].
[12] Kitamura, paragraph [0104].
[13] Kitamura, paragraph [0108].
[14] Kitamura, paragraph [0109].
[15] Kitamura, paragraph [0126].

remote video sources and the local video source is selected for recording. The record control is activated, and the selected video source is recorded into a file in response to the activation of the record control.[16]

Skarbo discloses that during a point-to-point data conference, typically each endpoint simultaneously transmits data, such as AV data, to the other. In a multipoint conference, typically an MCU it utilized to route data between various endpoints on a network such as a LAN. Such an MCU, because of communications and processing bandwidth constraints and other constraints, typically receives video streams from several of the endpoints and broadcasts only one of the video streams to all the endpoints, and mixes the audio streams received from the endpoints and transmits this mixed audio stream out to the endpoints. Such a decision of which video stream is to be broadcasted in the conference may be decided, for instance, by whichever endpoint user is currently speaking, or by a chair control who decides, as will be appreciated. Given sufficient processing and communications bandwidth capabilities, any number of video streams from the participants in the conference may be routed to the endpoints of the conference by the MCU.[17]

Internal Document: CUseeMe Networks Streaming Media Overview, March 15, 2001, by Richard Winefield (hereinafter Winefield). Winefield discloses that the CUseeMe Conference server (CUCS) sends data from a conference that has streaming enabled to a third party media encoder. The encoder in turn converts the video and audio from the conference into a file or live feed that is sent to a media server. The media server makes the broadcast available to users of third party viewers, such as Real Player, Windows Media Player, or IPTV viewer.[18] Winefield

[16] Skarbo, abstract.
[17] Skarbo, col. 4, lines 6-24.
[18] Winefield, page 3.

discloses using Media Encoder 4.x. Media Encoder 4.x allows a user to select which audio and video codecs he wants to broadcast with. CUCS can use any codec that is 8 kHz or less and mono.[19]

The power point presentation entitled *First Virtual Corporation, The Leader in Business Quality Video Networking* (hereinafter First Virtual) discusses intranet video collaboration.[20] First Virtual discloses separating bursty LAN data and streamed audio and video, which results in great performance of streamed traffic.[21] First Virtual discloses the use of an intrastream architecture that delivers business-quality video collaboration via a web-browser interface across a corporate intranet. First Virtual also discloses the use of MOS software to deliver the power of QOS to the intranet. The stream nature of video traffic requires the underlying capabilities of ATM for scalable implementation.[22]

The two page document entitled *Streaming Media, Product Overview, First Virtual Communications, Enables Interactive Conferences to be Streamed Live of Recorded* (hereinafter Streaming Media) discusses a feature that allows a user to capture video conferences and distribute content via popular streaming formats.[23] Streaming Media discloses a plurality of features on page 2, which include: allowing hundreds or thousands of participants to view a conference while it is happening, and recording and playing back a conference.[24]

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[19] Winefield, page 8.
[20] First Virtual, page 3.
[21] First Virtual, page 4.
[22] First Virtual, page 6.
[23] Streaming Media, page 1.
[24] Streaming Media, page 2.

The document entitled *Internet Telephony Product Reviews, MeetingPoint Conference Server, Version 4.0* (hereinafter MeetingPoint) discusses a software based virtual conferencing product that enables group meetings across IP networks. Users can share data, audio, and text with any H.323 standards-based client.[25]

MeetingPoint discloses that to host a conference, an administrator has to define the conferences. Unicast is a point-to-point connection where data is sent from one sender to one receiver, requiring a separate data stream for each recipient, while multicasting allows the administrator to send a single stream to a group of addresses.[26]

U.S. Patent No. 7,043,528 (Schmitt et al., hereinafter Schmitt) is a continuation of U.S. Application Serial No. 09/984,499 (U.S. Patent Publication No. 2002/0126201). These documents were both cited on the IDS filed July 28, 2006. Since these two applications share a common specification, they will be discussed together, with citations referring to U.S. Patent No. 7,043,528.

Schmitt is directed toward a system and method for connecting video conferencing to a distributed network. Schmitt discloses taking video and/or audio digital data content from a video conference session, which uses a video conference standard protocol technique, and distributes the video and/or audio digital data content onto a distributed network, which uses at least one multimedia streaming protocol technique.[27]

Fig. 6 of Schmitt is a flowchart outlining a method for distributing the audio and video content of a video conference as a multimedia data stream over a distributed network according to this invention. Beginning in step S100, operation

[25] MeetingPoint, page 1, and see also, page 3.
[26] MeetingPoint, page 4.
[27] Schmitt, col. 1, lines 19-25.

continues to step S200, where a video conference to be distributed as a multimedia data stream over distributed network is established between two or more video conference end point devices, if a peer-to-peer system is used, or between two or more video conference end point devices and a multipoint control unit. Next, in step S300, a video conference pseudo-participant end point unit according to this invention is connected to the established video conference. Then, in step S400, the digital video and audio streams of the video conference are supplied from the pseudo-participant end point unit to a streaming module. In step S500, the digital video and audio streams supplied to the streaming module are resupplied to one or more streaming servers that have one or more different protocols. Then, in step S600, each of the streaming servers converts the supplied digital video and audio streams provided to that particular streaming server into the corresponding protocol implemented by that streaming server. Next, in step S700, each different streaming server supplies the converted digital audio and video streams, now in the protocol corresponding to that particular streaming server, to one or more corresponding clients. In step S800, a determination is made whether the digital video and audio streams should continue to be captured from the video conference and supplied through the pseudo-participant end point and the streaming module to the streaming servers. If so, the operation jumps back to step S400. Otherwise, operation continues to step S900, where the method ends.[28]

The one page document entitled *CUseeMe Videoware, Rich Media Communications, First Virtual Communications* states "interfaces on the desktop through which Rich Media Communications can be delivered." Furthermore, the

[28] Schmitt, col. 10, line 37 to col. 11, line 25.

figure shows ways in which communications can extend beyond the desktop and the IP network.

EP 1069788 (Gray et al., hereinafter Gray) is directed toward a method of issuing passwords. Gray discloses that each password is associated with a specified unique entity. For example, an entity may be a site or switch in a communications network and the passwords may be required to enable a software patch to be installed on that switch. A database of details for each entity is stored and a manager is able to modify this database to indicate whether a password may be issued for a particular software patch and switch. A web interface is provided via which requests for passwords are made and an email engine provides email responses to the requests. Details of the requests are stored in the database and a manager is able to obtain information from the database using the web interface. This information enables the manager to control and monitor passwords that are issued and to detect misuse of the system or potential unauthorized password access.[29]

Fig. 13 of Gray shows a computer system for issuing passwords. The computer system includes: a memory 1301 arranged to store information about entities for which passwords may be issued; an input 1302 arranged to receive a request for a password associated with a given entity; and a processor 1303 arranged to issue the requested password request is made for an entity for which there is stored information indicating that a password may be issued.[30] The system also includes a web interface 1305, which is accessed by a user to request a password and enter information. A processor 1304 checks this information to determine whether a password may be issued.[31]

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[29] Gray, abstract.
[30] Gray, paragraph [0019].
[31] Gray, paragraph [0022].

The document entitled *User Guide, Version 5.0, CUseeMe Conference Server* (hereinafter User Guide) discusses a conference server that allows groups of users with networked personal computers or conferencing systems to interact in real-time, sharing any combination of audio, video, text and data.[32] User Guide discloses that when a user enables H.323 in a CUseeMe Conference Server conference, the user needs to specify which H.323 audio and video codecs are required for the conference. If the user does not designate codecs specifically, the CUseeMe conference server will automatically set the required audio codec to G.711 and the required video codec to H261.CIF.[33] Tables 4-2 and 4-3 list the audio and video codecs available on the Add h.323 Conference page.[34]

The CUseeMe Conference Server allows the user to specify the video and audio codec(s) that clients will use in a conference, which allows the user to choose what codecs are most appropriate for the conference he is creating.[35]

The document entitled *CUseeMe Conference Server, Telenet Interface Guide, Version 5.0* (hereinafter Interface Guide) provides a detailed description of each CUseeMe Conference Server Telnet command, as well as general information on using the commands.[36]

The "pref-aud-codecs" command allows the user to specify the use of certain audio codecs as either preferred or required in a conference. This allows the user to determine which audio codecs are most appropriate to the conference you are creating. If the user designates one or more codecs as preferred, it does not affect whether clients are allowed to connect to the conference. If the user does not specify

[32] User Guide, page 1-1.
[33] User Guide, page 2-17.
[34] User Guide, page 4-20.
[35] User Guide, page 4-26.
[36] Interface Guide, page 1-1.

any codecs for a CUseeMe-only conference, the CUseeMe Conference Server will allow users to connect to the conference using and codecs the choose. Some H.323 clients will allow the CUseeMe Conference Server to negotiate the correct codec, while others will not.[37]

The "pref-vid-codecs" command will allow the user to specify the use of certain video codecs as either preferred or required in a conference. This allows the user to choose which video codecs are most appropriate to the conference. CUseeMe client version 3.1 and later allows the CUseeMe Conference Server to negotiate with the correct codec with the client and allows the client to connect.[38]

WO 99/23560 (Ludwig et al., hereinafter Ludwig) is directed toward a networked multimedia system 10 that comprises a plurality of networks 40 and at least one storage server 100. A signal path interconnects the workstations 12 and the storage server 100. Each workstation 40 includes video and audio reproduction capabilities, as well as video and audio capture capabilities. Any given storage server 100 comprises a set of storage cells 120 that operate under the direction of a storage cell manager 160. A storage cell 120 may include one or more encoding 132 and transcoding converters configured to convert or transform audio and video signals originating at a workstation into a form suitable for storage. A storage cell 120 may further include one or more decoding converters 134 configured to convert stored signals into a form suitable for audio and video signal reproduction at a workstation. Each storage cell 120 additionally includes at least one storage device 150 and storage device controller 152 capable of storing, for later retrieval, signals generated by one or more converters. The storage cell controller responds to signals received

[37] Interface Guide, page 2-92.
[38] User Interface, page 2-97.

from the workstations 40, and oversees the operation of the storage cells to facilitate the storage of converted audio and video signals in at least one file that can be simultaneously accessed by one or more application programs executing on one or more workstations.[39]

Fig. 3 of Ludwig shows a block diagram of a Collaborative Multimedia Computing Environment (CMCE) employing an audio/video server system (AVSS). Fig. 3 shows CMC) 10 employing an AVSS 100 constructed in accordance with the present invention. The CMCE 10 comprises a data network 20; an A/V network 30; a plurality of user workstations 40 and/or a set of A/V conference rooms 45; an Audio/Video Server System (AVSS) 100; and a set of supporting server systems that include an e-mail system 50, an intranet server system 60, and a firewall/internet gateway system 70. The data network 20 couples the A/V network 30, the workstations 40, the A/V conference room (s) 45, each supporting server system 50,60,70, and the AVSS 100. The data network 20 also maintains a wide area link that is coupled to a Wide Area Network (WAN) gateway 25, which is coupled to a first WAN 29. The A/V network 30 couples the workstations 40, the A/V conference rooms 45, and the AVSS 100. The A/V network 30 maintains at least one trunk line coupling 16 to a remote and/or another local A/V network 30. The A/V network 30 additionally maintains a trunk line coupling 16 to a coder/decoder (codec) gateway 38, which is coupled to a second WAN 39.[40]

Each AVSC 120 serves as an A/V file repository, as well as a repository for shared A/V processing resources, including encoders, decoders, and possibly transcoders.[41] The AVSC object memory 152 stores a plurality of AVSC software

[39] Ludwig, abstract.
[40] Ludwig, page 11, lines 1-15.
[41] Ludwig, page 18, lines 19-24.

objects that direct AVSC hardware allocation and resource locking; A/V file encoding, decoding, and transcoding operations; and file management operations such as file replication, transfer, and deletion, as described in detail below. The AVSC objects also maintain the contents of the AVSC state memory 154, which includes the following information: 1) encoder/decoder/transcoder resource capability and current status/utilization; 2) current storage device capacity and utilization; 3) a time-stamped and indexed request queue, for both incoming and outgoing requests; 4) a time-stamped and indexed message queue, for both incoming and outgoing messages; 5) a time-stamped and indexed file transfer event queue; and 6) an AVSC event log, specifying standard time-stamped events, as well as occurrences of encoder, decoder, transcoder, storage device, and/or network faults.[42]

The above-noted references do not teach or suggest alone, or in proper combination, at least,

> applying a selected encoding process to a received audio visual signal, said encoding process being selected depending on the contents of said at least one protocol signal read

of Claim 37,

> encoding a keyframe into the encoded output at the same time position at which the keyframe was detected in the original received audio visual signal.

of Claim 63,

> encoding an index marker at the time position at which the content switch was detected.

of Claim 64,

[42] Ludwig, page 20, lines 1-16.

encoding an index marker at the same time position or adjacent to the time position of the keyframe encoded.

of Claim 66,

time compressing the encoded output content during the time period in which said low content state is detected within the videoconference transmission received

of Claim 67, and

(iii) receiving encoding preferences from at least one user,

(iv) selecting from a set of encoding processes a subset of encoding processes which can be implemented using the user's preferences and the encoding characteristics, and

(v) displaying the subset of encoding processes to a user.

of Claim 69.

In view of the above-noted distinctions, Applicants respectfully submit that Claims 37 (and dependent Claims 38-62), 63, 64 (and dependent Claim 65), 66, 67 (and dependent Claim 68), and 69 patentably distinguish over the references of record.

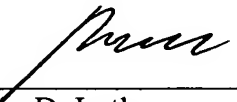
III. Conclusion

The petition to make special meets all the requirements of MPEP §708.02(VIII), and therefore, should be granted. Accordingly, Applicants respectfully request that this Application be advanced out of turn for examination, and that the assigned Examiner,

pursuant to the suggestions of MPEP §708.02(VIII), contact the undersigned to schedule an interview for advancing the prosecution of this case.

Respectfully submitted,

OBLON, SPIVAK, McCLELLAND,
MAIER & NEUSTADT, P.C.



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